Claims:

1. A method suitable for use in reducing echo in a communication system, said method comprising:
    a) receiving a first signal including a voice component, the voice component being associated to a speaker;
    b) receiving a second signal including an echo component, the echo component being correlated to the first signal;
    c) processing said first signal to derive an estimate of a harmonic feature of the voice component;
    d) processing said second signal at least in part on the basis of the harmonic component of the voice component to remove at least in part the echo component such as to derive an echo reduced signal;
    e) releasing the echo reduced signal.

2. A method as defined in claim 1, wherein the harmonic feature of the voice component is an estimate of the pitch associated to the voice component.

3. A method as defined in claim 2, said method comprising applying a filtering operation to the second signal at least in part on the basis of the estimate of the pitch associated to the voice component to derive the echo reduced signal.

4. A method as defined in claim 2, wherein the voice component is a first voice component, said second signal including a second voice component, the second voice component being associated to a second speaker.

5. A method as defined in claim 4, said method comprising:
    i. generating a set of filter coefficients at least in part on the basis of the estimate of the pitch associated to the first voice component;

ii. applying a filtering operation to the second signal on the basis of the set of filter coefficients generated in i. to derive the echo reduced signal.

6. A method as defined in claim 5, said method comprising:
   a) processing the first signal and the second signal to detect an occurrence of double talk;
   b) in response to detection of an occurrence of double talk, processing the estimate of the pitch associated to the first voice component to derive a number of filter coefficients in said set of filter coefficients.

7. A method as defined in claim 5, wherein said filtering operation is a finite-impulse response (FIR) filtering operation.

8. A method as defined in claim 7, wherein said filtering operation is an asymmetric filtering operation.

9. A method as defined in claim 5, wherein said filtering operation is adapted to filter a set of frequencies from said second signal to derive the echo reduced signal, the set of frequencies including frequencies which are integer multiples of the estimate of the pitch associated to the first voice component.

10. An apparatus suitable for use in reducing echo in a communication system, said apparatus comprising:
    a) a first input for receiving a first signal including a voice component, the voice component being associated to a speaker;
    b) a second input for receiving a second signal including an echo component, the echo component being correlated to the first signal;
    c) a processing unit in communication with said first input and said second input, said processing unit being operative for:
    i. processing said first signal to derive a harmonic feature of the voice component;

ii. processing said second signal at least in part on the basis of the harmonic feature of the voice component to remove at least in part the echo component such as to derive an echo reduced signal;

d) an output for releasing the echo reduced signal.

11. An apparatus as defined in claim 10, wherein the harmonic feature of the voice component is an estimate of the pitch associated to the voice component.

12. An apparatus as defined in claim 11, wherein said processing unit is adapted for applying a filtering operation to the second signal at least in part on the basis of the estimate of the pitch associated to the voice component to derive the echo reduced signal.

13. An apparatus as defined in claim 11, wherein the voice component is a first voice component, said second signal including a second voice component, the second voice component being associated to a second speaker.

14. An apparatus as defined in claim 13, wherein said processing unit is adapted for:

i. generating a set of filter coefficients at least in part on the basis of the estimate of the pitch associated to the first voice component;

ii. applying a filtering operation to the second signal on the basis of the set of filter coefficients generated in i. to derive the echo reduced signal.

15. An apparatus as defined in claim 14, wherein said processing unit is adapted for:

a) processing the first signal and the second signal to detect an occurrence of double talk;

b) in response to detection of an occurrence of double talk, processing the estimate of the pitch associated to the first voice component to derive a number of filter coefficients in said set of filter coefficients.

16. An apparatus as defined in claim 14, wherein said filtering operation is a finite-impulse response (FIR) filtering operation.

17. An apparatus as defined in claim 14, wherein said filtering operation is an asymmetric filtering operation.

18. An apparatus as defined in claim 14, wherein said filtering operation is adapted to filter a set of frequencies from said second signal to derive the echo reduced signal, the set of frequencies including frequencies which are integer multiples of the estimate of the pitch associated to the first voice component.

19. An apparatus as defined in claim 11, wherein said processing unit comprises:
   a) a pitch detection unit operative for processing the first signal to derive the estimate of a pitch associated to the voice component;
   b) a filter adaptation module operative for generating a set of filter coefficients at least in part on the basis of the estimate of a pitch associated to the voice component;
   c) a filter unit operative for processing the second signal on the basis of the set of filter coefficients to derive the echo reduced signal.

20. An apparatus as defined in claim 19, wherein said filter unit is a finite impulse response (FIR) filter.

21. An apparatus as defined in claim 20, wherein said filter unit is an asymmetric filter.

22. A computer readable medium including a program element suitable for execution by a computing apparatus for use in reducing echo in a communication system, said computing apparatus comprising:
   a) a memory unit;
   b) a processor operatively connected to said memory unit, said program element when executing on said processor being operative for:

i. receiving a first signal including a voice component, the voice component being associated to a speaker;

ii. receiving a second signal including an echo component, the echo component being correlated to the first signal;

iii. processing said first signal to derive a harmonic feature of the voice component;

iv. processing said second signal at least in part on the basis of harmonic feature of the voice component to remove at least in part the echo component such as to derive an echo reduced signal;

v. releasing the echo reduced signal.

23. A computer readable medium as defined in claim 22, wherein the harmonic feature of the voice component is an estimate of the pitch associated to the voice component.

24. A computer readable medium as defined in claim 23, said program element when executing on said processor being operative for applying a filtering operation to the second signal at least in part on the basis of the estimate of the pitch associated to the voice component to derive the echo reduced signal.

25. A computer readable medium as defined in claim 23, wherein the voice component is a first voice component, said second signal including a second voice component, the second voice component being associated to a second speaker.

26. A computer readable medium as defined in claim 25, said program element when executing on said processor being operative for:

i. generating a set of filter coefficients at least in part on the basis of the estimate of the pitch associated to the first voice component;

ii. applying a filtering operation to the second signal on the basis of the set of filter coefficients generated in i. to derive the echo reduced signal.

27. A computer readable medium as defined in claim 26, wherein the filtering operation is characterized by a set of filter coefficients.

28. A computer readable medium as defined in claim 27, said program element when executing on said processor being responsive detection of an occurrence of double talk for processing the estimate of the pitch associated to the first voice component to derive a number of filter coefficients in said set of filter coefficients.

29. A computer readable medium as defined in claim 26, wherein said filtering operation is a finite-impulse response (FIR) filtering operation.

30. A computer readable medium as defined in claim 29, wherein said filtering operation is an asymmetric filtering operation.

31. A computer readable medium as defined in claim 26, wherein said filtering operation is adapted to filter a set of frequencies from said second signal to derive the echo reduced signal, the set of frequencies including frequencies which are integer multiples of the estimate of the pitch associated to the first voice component.

32. A computer readable medium as defined in claim 23, wherein said program element when executing on said processor being operative for implementing:

a) a pitch detection unit operative for processing the first signal to derive the estimate of a pitch associated to the voice component;

b) a filter adaptation module operative for generating a set of filter coefficients at least in part on the basis of the estimate of a pitch associated to the voice component;

c) a filter unit operative for processing the second signal on the basis of the set of filter coefficients to derive the echo reduced signal.

33. A filter adaptation apparatus suitable for generating a set of filter coefficients, said filter adaptation apparatus comprising:

a) an input for receiving an estimate of a harmonic feature of a voice component in a signal;
b) a processing unit operative for generating a set of filter coefficients at least in part on the basis of the harmonic feature of the voice component;
c) an output for releasing the set of filter coefficients for use by a filter unit.

34. A filter adaptation apparatus as defined in claim 33, wherein the harmonic feature of the voice component is an estimate of the pitch associated to the voice component.

35. A filter adaptation apparatus as defined in claim 34, wherein said set of filter coefficients forms a finite-impulse response filter.

36. A filter adaptation apparatus as defined in claim 35, wherein said set of filter coefficients provide a symmetric impulse response filter.

37. A filter adaptation apparatus as defined in claim 34, wherein said processing unit operative for:
a) generating a first set of filter coefficients at least in part on the basis of the estimate of a pitch associated to the voice component, the first set of filter coefficients providing a symmetric impulse response filter;
b) processing the first set of filter coefficients to generate a second set of filter coefficients, the second set of filter coefficients providing an asymmetric impulse response filter.

38. A filter adaptation apparatus as defined in claim 37, wherein the second set of filter coefficients is derived by shifting the first set of filter coefficient.

39. A filter adaptation apparatus as defined in claim 38, wherein the first set of filter coefficients includes N coefficients, the second set of filter coefficients being derived by shifting the first set of filter coefficients by N/2.

40. A filter adaptation apparatus as defined in claim 34, wherein said input is a first input and said signal is a first signal, said filter adaptation apparatus further comprising:
    a) a second input for receiving a double talk indicator indicative of an occurrence of double talk in a second signal, the second signal including an echo component correlated with the first signal;
    b) said processing unit being responsive to a double talk indicator indicative of an occurrence of double talk for generating the set of filter coefficients.

41. An apparatus suitable for use in reducing echo in a communication system, said apparatus comprising:
    a) means for receiving a first signal including a voice component, the voice component being associated to a speaker;
    b) means for receiving a second signal including an echo component, the echo component being correlated to the first signal;
    c) means for processing said first signal to derive a harmonic feature of the voice component;
    d) means for processing said second signal at least in part on the basis of the harmonic feature of the voice component to remove at least in part the echo component such as to derive an echo reduced signal;
    e) means for releasing the echo reduced signal.

42. A filter adaptation apparatus suitable for generating a set of filter coefficients, said filter adaptation apparatus comprising:
    a) an input for receiving a harmonic feature of a time varying signal;
    b) a processing unit operative for generating a set of filter coefficients at least in part on the basis of the harmonic feature, said set of filter coefficients defining a finite-impulse response filter;
    c) an output for releasing the set of filter coefficients for use by a filter unit.

43. A filter adaptation apparatus as defined in claim 42, wherein said processing unit operative for:

a) generating a first set of filter coefficients at least in part on the basis of the estimate of the harmonic feature, the first set of filter coefficients providing a symmetric impulse response filter;

b) processing the first set of filter coefficients to generate a second set of filter coefficients, the second set of filter coefficients providing an asymmetric impulse response filter.

44. A filter adaptation apparatus as defined in claim 43, wherein the second set of filter coefficients is derived by performing a circular shift of the first set of filter coefficient.

45. A filter adaptation apparatus as defined in claim 44, wherein the first set of filter coefficients includes N coefficients, the second set of filter coefficients being derived by shifting the first set of filter coefficients by N/2.

46. A filter adaptation apparatus as defined in claim 42, wherein said finite-impulse response filter is characterized by a length N, said finite-impulse response filter having a delay that is less than N/2 for certain frequencies bands.

47. A filter adaptation apparatus as defined in claim 42, wherein said finite-impulse response filter is characterized by no delay for certain frequencies bands.

48. A filter adaptation apparatus as defined in claim 42, wherein said time varying signal is a voice signal and said harmonic feature is indicative of an estimate of the pitch associated with the voice signal.